(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
3 May 2001 (03.05.2001)

PCT

(10) International Publication Number
WO 01/31816 A1

(51) International Patent Classification[7]: H04H 1/00, 9/00

(21) International Application Number: PCT/US00/03829

(22) International Filing Date: 14 February 2000 (14.02.2000)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
09/428,425 27 October 1999 (27.10.1999) US

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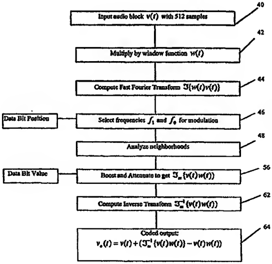
(81) Designated States *(national)*: AE, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, CA, CH, CN, CR, CU, CZ, DE, DK, DM, EE, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, TZ, UA, UG, UZ, VN, YU, ZA, ZW.

(84) Designated States *(regional)*: ARIPO patent (GH, GM, KE, LS, MW, SD, SL, SZ, TZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG).

Published:
— *With international search report.*

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

(54) Title: SYSTEM AND METHOD FOR ENCODING AN AUDIO SIGNAL FOR USE IN BROADCAST PROGRAM IDENTIFICATION SYSTEMS, BY ADDING INAUDIBLE CODES TO THE AUDIO SIGNAL



(57) Abstract: Blocks of audio are encoded based upon corresponding first and second frequencies. The first and second frequencies are hopped from block to block. An audio quality measure (AQM) is computed for each block of audio such that, if x out of y blocks of audio have an AQM greater than a first predetermined threshold, encoding is suspended. For example, x may be nine and y may be 16. Also, if a ratio of the energy in a front part of a block of audio to the energy in a rear part of the block of audio is greater than a second predetermined threshold, that block of audio is not encoded even though x out of y blocks of audio have an AQM greater than the first predetermined threshold. Multiple distributors of the audio may encode the audio with their corresponding identities using the above processes.

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SYSTEM AND METHOD FOR ENCODING AN AUDIO SIGNAL FOR USE IN BROADCAST PROGRAM IDENTIFICATION SYSTEMS, BY ADDING INAUDIBLE CODES TO THE AUDIO SIGNAL

Related Application

This application is a continuation-in-part of U.S. Patent Application Serial No. 09/116,397 filed July 16, 1998. This application also contains disclosure similar to the disclosure in U.S. Application Serial No. (28019/35519).

Technical Field of the Invention

The present invention relates to spectral audio encoding useful, for example, in modulating broadcast signals in order to add identifying codes thereto.

Background of the Invention

Several approaches to metering the video and/or audio tuned by television and/or radio receivers in order to determine the sources or identities of corresponding television or radio programs are known. For example, one approach is to real time correlate a program to which a receiver is tuned with each of the programs available to the receiver. An apparatus useful for this measurement approach is found in the teachings of Lu et al. in U.S. Patent No. 5,594,934.

Another approach is to extract a characteristic signature (or a characteristic signature set) from the program selected for viewing and/or listening, and to compare the characteristic signature (or characteristic signature set) with reference signatures (or reference signature sets) collected from known transmission sources at a reference site.

Although the reference site could be the viewer's household, the reference site is usually at a location which is remote from the households of all of the viewers being monitored. Systems using signature extraction are taught by Lert and Lu in U.S. Patent No. 4,677,466 and by Kiewit and Lu in U.S. Patent No. 4,697,209.

In signature extraction systems, audio characteristic signatures are often utilized. Typically, these characteristic signatures are extracted by a unit located at the monitored receiver, sometimes referred to as a site unit. The site unit monitors the audio output of a television or radio receiver either by means of a microphone that picks up the sound from the speakers of the monitored receiver or by means of an output line from the monitored receiver. The site unit extracts and transmits the characteristic signatures to a central household unit, sometimes referred to as a home unit. Each characteristic signature is designed to uniquely characterize the audio signal tuned by the receiver during the time of signature extraction.

Characteristic signatures are typically transmitted from the home unit to a central office where a matching operation is performed between the characteristic signatures and a set of reference signatures extracted at a reference site from all of the audio channels that could have been tuned by the receiver in the household being monitored. A matching score

is computed by a matching algorithm and is used to determine the identity of the program to which the monitored receiver was tuned or the program source (such as a broadcaster) of the tuned program.

Yet another approach to metering video and/or audio tuned by televisions and/or radios is to add ancillary identification codes to television and/or radio programs and to detect and decode the ancillary codes in order to identify the encoded programs or the corresponding program sources when the programs are tuned by monitored receivers. There are many arrangements for adding an ancillary code to a signal in such a way that the added code is not noticed. It is well known in television broadcasting, for example, to hide such ancillary codes in non-viewable portions of video by inserting them into either the video's vertical blanking interval or horizontal retrace interval. An exemplary system which hides codes in non-viewable portions of video is referred to as "AMOL" and is taught in U.S. Patent No. 4,025,851. This system is used by the assignee of this application for monitoring transmissions of television programming as well as the times of such transmissions.

Other known video encoding systems have sought to bury the ancillary code in a portion of a television signal's transmission bandwidth that otherwise carries little signal energy. An example of such a system is disclosed by Dougherty

in U.S. Patent No. 5,629,739, which is assigned to the assignee of the present application.

Other methods and systems add ancillary codes to audio signals for the purpose of identifying the signals and, perhaps, for tracing their courses through signal distribution systems. Such arrangements have the obvious advantage of being applicable not only to television, but also to radio transmissions and to pre-recorded music. Moreover, ancillary codes which are added to audio signals may be reproduced in the audio signal output by a speaker. Accordingly, these arrangements offer the possibility of non-intrusively intercepting and decoding the codes with equipment that has a microphone as an input. In particular, these arrangements provide an approach to measuring program audiences by the use of portable metering equipment carried by panelists.

One such audio encoding system is disclosed by Crosby, in U.S. Patent No. 3,845,391. In this system, a code is inserted in a narrow frequency "notch" from which the original audio signal is deleted. The notch is made at a fixed predetermined frequency (e.g., 40 Hz). This approach led to codes that were audible when the original audio signal containing the code was of low intensity.

A series of improvements followed the Crosby patent. Thus, Howard, in U.S. Patent No. 4,703,476, teaches the use of two separate notch frequencies for the mark and the space

portions of a code signal. Kramer, in U.S. Patent No. 4,931,871 and in U.S. Patent No. 4,945,412 teaches, *inter alia*, using a code signal having an amplitude that tracks the amplitude of the audio signal to which the code is added.

Program audience measurement systems in which panelists are expected to carry microphone-equipped audio monitoring devices that can pick up and store inaudible codes transmitted in an audio signal are also known. For example, Aijalla et al., in WO 94/11989 and in U.S. Patent No. 5,579,124, describe an arrangement in which spread spectrum techniques are used to add a code to an audio signal so that the code is either not perceptible, or can be heard only as low level "static" noise. Also, Jensen et al., in U.S. Patent No. 5,450,490, teach an arrangement for adding a code at a fixed set of frequencies and using one of two masking signals, where the choice of masking signal is made on the basis of a frequency analysis of the audio signal to which the code is to be added. Jensen et al. do not teach a coding arrangement in which the code frequencies vary from block to block. The intensity of the code inserted by Jensen et al. is a predetermined fraction of a measured value (e.g., 30 dB down from peak intensity) rather than comprising relative maxima or minima.

Moreover, Preuss et al., in U.S. Patent No. 5,319,735, teach a multi-band audio encoding arrangement in which a spread spectrum code is inserted in recorded music at

a fixed ratio to the input signal intensity (code-to-music ratio) that is preferably 19 dB. Lee et al., in U.S. Patent No. 5,687,191, teach an audio coding arrangement suitable for use with digitized audio signals in which the code intensity is made to match the input signal by calculating a signal-to-mask ratio in each of several frequency bands and by then inserting the code at an intensity that is a predetermined ratio of the audio input in that band. As reported in this patent, Lee et al. have also described a method of embedding digital information in a digital waveform in pending U.S. application Serial No. 08/524,132.

It will be recognized that, because ancillary codes are preferably inserted at low intensities in order to prevent the code from distracting a listener of program audio, such codes may be vulnerable to various signal processing operations. For example, although Lee et al. discuss digitized audio signals, it may be noted that many of the earlier known approaches to encoding an audio signal are not compatible with current and proposed digital audio standards, particularly those employing signal compression methods that may reduce the signal's dynamic range (and thereby delete a low level code) or that otherwise may damage an ancillary code. In this regard, it is particularly important for an ancillary code to survive compression and subsequent de-compression by the AC-3 algorithm or by one of the algorithms recommended in the

ISO/IEC 11172 MPEG standard, which is expected to be widely used in future digital television transmission and reception systems.

U.S. Patent Application Serial No. 09/116,397 filed July 16, 1998 discloses a system and method for inserting a code into an audio signal so that the code is likely to survive compression and decompression as required by current and proposed digital audio standards. In this system and method, spectral modulation at selected code frequencies is used to insert the code into the audio signal. These code frequencies are varied from audio block to audio block, and the spectral modulation may be implemented as amplitude modulation, modulation by frequency swapping, phase modulation, and/or odd/even index modulation.

In most audio signals of the type used in television systems, a code inserted by spectral modulation in accordance with the aforementioned patent application is substantially inaudible. However, there are some instances where the code may be undesirably audible. The present invention addresses one or more of these instances. The present application also addresses methods of multi-level coding.

Summary of the Invention

According to one aspect of the present invention, a method for encoding first and second blocks of audio with

corresponding first and second binary code bits comprises the following steps: a) selecting first and second frequencies from a frequency spectrum of the first block of audio; b) modulating the audio based upon the first and second frequencies to thereby encode the first block of audio with the first binary code bit; c) selecting third and fourth frequencies from a frequency spectrum of the second block of audio, wherein the third and fourth frequencies bear a predetermined offset relationship to the first and second frequencies; and, d) modulating the audio based upon the third and fourth frequencies to thereby encode the second block of audio with the second binary code bit.

According to another aspect of the present invention, a method for encoding a block of audio with a binary code bit comprises the following steps: a) selecting a frequency from a frequency spectrum of the block of audio; b) selectively amplifying an odd index frequency in a neighborhood of the selected frequency to be a local maximum if the block of audio is to be encoded with the binary code bit having a first value; and, c) selectively amplifying an even index frequency in a neighborhood of the selected frequency to be a local maximum if the block of audio is to be encoded with the binary code bit having a second value.

According to still another aspect of the present invention, a method for encoding blocks of audio with binary

code bits comprises the following steps: a) determining an audio quality measure AQM for each block of audio; b) comparing the AQM corresponding to each block of audio to AQM_{THRESH}, wherein AQM_{THRESH} is a predetermined audio quality measure reference; c) if AQM < AQM_{THRESH} for x blocks of audio out of y blocks of audio, encoding the blocks of audio with binary bits, wherein x and y are corresponding predetermined numbers of blocks of audio; and, d) if AQM > AQM_{THRESH} for the x blocks of audio out of the y blocks of audio, suspending encoding of the blocks of audio.

According to yet another aspect of the present invention, a method is provided to encode a block of audio with a binary code bit. The block of audio has an energy. The method comprises the following steps: a) determining a ratio E_1/E_2, wherein E_1 is the energy in a first portion of the block of audio, and wherein E_2 is the energy in a second portion of the block of audio; b) modulating the block of audio with the binary code bit if $E_1/E_2 > E_{PRE}$, wherein E_{PRE} is a predetermined reference; and, c) not modulating the block of audio with the binary code bit if $E_1/E_2 < E_{PRE}$.

According to a further aspect of the present invention, a method of encoding blocks of audio with binary code bits comprises the following steps: a) encoding each of the blocks of audio with a binary bit by modulating the audio within the corresponding block of audio at selected first and

second frequencies, wherein the selected first and second frequencies are hopped from block to block; and, b) executing step a) so as to indicate first and second levels of distribution of the audio.

According to a still further aspect of the present invention, a method for decoding first and second blocks of audio in order to recover corresponding first and second binary code bits therefrom comprises the following steps: a) detecting first and second frequencies from a frequency spectrum of the first block of audio; b) demodulating the first and second frequencies in order to recover to the first binary code bit; c) detecting third and fourth frequencies from a frequency spectrum of the second block of audio, wherein the third and fourth frequencies bear a predetermined offset relationship to the first and second frequencies; and, d) demodulating the third and fourth frequencies in order to recover the second binary code bit.

According to yet a further aspect of the present invention, a method of decoding blocks of audio encoded with binary code bits comprises the following steps: a) decoding each of the blocks of audio in order to recover a corresponding binary bit by demodulating the audio within the corresponding block of audio at selected first and second frequencies, wherein the selected first and second frequencies are

hopped from block to block; and, b) executing step a) so as identify first and second distributors of the audio.

According to another aspect of the present invention, a method of decoding a block of audio in order to recover a binary code bit therefrom comprises the following steps: a) detecting a frequency having an amplitude maximum within a selected frequency neighborhood of the block of audio; b) if the frequency detected in step a) corresponds to an odd frequency index, decoding the frequency as a binary code bit having a first value; and, c) if the frequency detected in step a) corresponds to an even frequency index, decoding the frequency as a binary code bit having a second value.

Brief Description of the Drawing

These and other features and advantages will become more apparent from a detailed consideration of the invention when taken in conjunction with the drawings in which:

Figure 1 is a schematic block diagram of an audience measurement system employing the signal coding and decoding arrangements of the present invention;

Figure 2 is flow chart depicting steps performed by an encoder of the system shown in Figure 1;

Figure 3 is a spectral plot of an audio block, wherein the thin line of the plot is the spectrum of the

original audio signal and the thick line of the plot is the spectrum of the signal modulated in accordance with the present invention;

Figure 4 depicts a window function which may be used to prevent transient effects that might otherwise occur at the boundaries between adjacent encoded blocks;

Figure 5 is a schematic block diagram of an arrangement for generating a seven-bit pseudo-noise synchronization sequence;

Figure 6 is a spectral plot of a "triple tone" audio block which forms the first block of a preferred synchronization sequence, where the thin line of the plot is the spectrum of the original audio signal and the thick line of the plot is the spectrum of the modulated signal;

Figure 7a schematically depicts an arrangement of synchronization and information blocks usable to form a complete code message;

Figure 7b schematically depicts further details of the synchronization block shown in Fig. 7a;

Figure 8 is a flow chart depicting steps performed by a decoder of the system shown in Figure 1; and,

Figure 9 illustrates an encoding arrangement in which audio encoding delays are compensated in the video data stream.

Detailed Description of the Invention

Audio signals are usually digitized at sampling rates that range between thirty-two kHz and forty-eight kHz. For example, a sampling rate of 44.1 kHz is commonly used during the digital recording of music. However, digital television ("DTV") is likely to use a forty eight kHz sampling rate. Besides the sampling rate, another parameter of interest in digitizing an audio signal is the number of binary bits used to represent the audio signal at each of the instants when it is sampled. This number of binary bits can vary, for example, between sixteen and twenty four bits per sample. The amplitude dynamic range resulting from using sixteen bits per sample of the audio signal is ninety-six dB. This decibel measure is the ratio between the square of the highest audio amplitude (2^{16} = 65536) and the lowest audio amplitude (1^2 = 1). The dynamic range resulting from using twenty-four bits per sample is 144 dB. Raw audio, which is sampled at the 44.1 kHz rate and which is converted to a sixteen-bit per sample representation, results in a data rate of 705.6 kbits/s.

Compression of audio signals is performed in order to reduce this data rate to a level which makes it possible to transmit a stereo pair of such data on a channel with a throughput as low as 192 kbits/s. This compression typically is accomplished by transform coding. A block consisting of N_d = 1024 samples, for example, may be decomposed, by application

of a Fast Fourier Transform or other similar frequency analysis process, into a spectral representation. In order to prevent errors that may occur at the boundary between one block and the previous or subsequent block, overlapped blocks are commonly used. In one such arrangement where 1024 samples per overlapped block are used, a block includes 512 samples of "old" samples (i.e., samples from a previous block) and 512 samples of "new" or current samples. The spectral representation of such a block is divided into critical bands where each band comprises a group of several neighboring frequencies. The power in each of these bands can be calculated by summing the squares of the amplitudes of the frequency components within the band.

Audio compression is based on the principle of masking that, in the presence of high spectral energy at one frequency (i.e., the masking frequency), the human ear is unable to perceive a lower energy signal if the lower energy signal has a frequency (i.e., the masked frequency) near that of the higher energy signal. The lower energy signal at the masked frequency is called a masked signal. A masking threshold, which represents either (i) the acoustic energy required at the masked frequency in order to make it audible or (ii) an energy change in the existing spectral value that would be perceptible, can be dynamically computed for each band. The frequency components in a masked band can be represented in a

coarse fashion by using fewer bits based on this masking threshold. That is, the masking thresholds and the amplitudes of the frequency components in each band are coded with a smaller number of bits which constitute the compressed audio. Decompression reconstructs the original signal based on this data.

Figure 1 illustrates an audience measurement system 10 in which an encoder 12 adds an ancillary code to an audio signal portion 14 of a program signal to be transmitted. Alternatively, the encoder 12 may be provided, as is known in the art, at some other location in the program signal distribution chain. A transmitter 16 transmits the encoded audio signal portion with a video signal portion 18 of the program signal. When the encoded signal is received by a receiver 20 located at a statistically selected metering site 22, the ancillary code is recovered by processing the audio signal portion of the received program signal even though the presence of that ancillary code is imperceptible to a listener when the encoded audio signal portion is supplied to speakers 24 of the receiver 20. To this end, a decoder 26 is connected either directly to an audio output 28 available at the receiver 20 or to a microphone 30 placed in the vicinity of the speakers 24 through which the audio is reproduced. The received audio signal can be either in a monaural or stereo format.

ENCODING BY SPECTRAL MODULATION

In order for the encoder 12 to embed a digital code in an audio data stream in a manner compatible with compression technology, the encoder 12 should preferably use frequencies and critical bands that match those used in compression. The block length N_C of the audio signal that is used for coding may be chosen such that, for example, $jN_C = N_d = 1024$, where j is an integer. A suitable value for N_C may be, for example, 512. As depicted by a step 40 of the flow chart shown in Figure 2, which is executed by the encoder 12, a first block v(t) of N_C samples is derived from the audio signal portion 14 by the encoder 12 such as by use of an analog to digital converter, where v(t) is the time-domain representation of the audio signal within the block. An optional window may be applied to v(t) at a block 42 as discussed below in additional detail. Assuming for the moment that no such window is used, a Fourier Transform $\Im\{v(t)\}$ of the block v(t) to be coded is computed at a step 44. (The Fourier Transform implemented at the step 44 may be a Fast Fourier Transform.)

The frequencies resulting from the Fourier Transform are indexed in the range -256 to +255, where an index of 255 corresponds to exactly half the sampling frequency f_S. Therefore, for a forty-eight kHz sampling frequency, the highest index would correspond to a frequency of twenty-four kHz.

Accordingly, for purposes of this indexing, the index closest to a particular frequency component f_j resulting from the Fourier Transform $\Im\{v(t)\}$ is given by the following equation:

$$I_j = (\frac{255}{24}) \cdot f_j \qquad (1)$$

where equation (1) is used in the following discussion to relate a frequency f_j and its corresponding index I_j.

The code frequencies f_i used for coding a block may be chosen from the Fourier Transform $\Im\{v(t)\}$ at a step 46 in the 4.8 kHz to 6 kHz range in order to exploit the higher auditory threshold in this band. Also, each successive bit of the code may use a different pair of code frequencies f_1 and f_0 denoted by corresponding code frequency indexes I_1 and I_0. There are two preferred ways of selecting the code frequencies f_1 and f_0 at the step 46 so as to create an inaudible wide-band noise like code.

(a) Direct Sequence

One way of selecting the code frequencies f_1 and f_0 at the step 46 is to compute the code frequencies by use of a frequency hopping algorithm employing a hop sequence H_S and a shift index I_{shift}. For example, if N_s bits are grouped to-

gether to form a pseudo-noise sequence, H_S is an ordered sequence of N_s numbers representing the frequency deviation relative to a predetermined reference index I_{5k}. For the case where $N_s = 7$, a hop sequence $H_S = \{2,5,1,4,3,2,5\}$ and a shift index $I_{shift} = 5$, for example, could be used. In general, the indices for the N_s bits resulting from a hop sequence may be given by the following equations:

$$I_1 = I_{5k} + H_s - I_{shift} \tag{2}$$

and

$$I_0 = I_{5k} + H_s + I_{shift}. \tag{3}$$

One possible choice for the reference frequency f_{5k} is five kHz, for example, which corresponds to a predetermined reference index $I_{5k} = 53$. This value of f_{5k} is chosen because it is above the average maximum sensitivity frequency of the human ear. When encoding a first block of the audio signal, I_1 and I_0 for the first block are determined from equations (2) and (3) using a first of the hop sequence numbers; when encoding a second block of the audio signal, I_1 and I_0 for the second block are determined from equations (2) and (3) using a second of the hop sequence numbers; and so on. For the fifth bit in the sequence $\{2,5,1,4,3,2,5\}$, for example, the hop sequence

value is three and, using equations (2) and (3), produces an index $I_1 = 51$ and an index $I_0 = 61$ in the case where $I_{shift} = 5$. In this example, the mid-frequency index is given by the following equation:

$$I_{mid} = I_{5k} + 3 = 56 \qquad (4)$$

where I_{mid} represents an index mid-way between the code frequency indices I_1 and I_0. Accordingly, each of the code frequency indices is offset from the mid-frequency index by the same magnitude, I_{shift}, but the two offsets have opposite signs.

(b) Hopping based on low frequency maximum

Another way of selecting the code frequencies at the step 46 is to determine a frequency index I_{max} at which the spectral power of the audio signal, as determined as the step 44, is a maximum in the low frequency band extending from zero Hz to two kHz. In other words, I_{max} is the index corresponding to the frequency having maximum power in the range of 0 - 2 kHz. It is useful to perform this calculation starting at index 1, because index 0 represents the "local" DC component and may be modified by high pass filters used in compression. The code frequency indices I_1 and I_0 are chosen relative to the frequency index I_{max} so that they lie in a higher frequency band at which the human ear is relatively less sensitive.

Again, one possible choice for the reference frequency f_{5k} is five kHz corresponding to a reference index $I_{5k} = 53$ such that I_1 and I_0 are given by the following equations:

$$I_1 = I_{5k} + I_{max} - I_{shift} \qquad (5)$$

and

$$I_0 = I_{5k} + I_{max} + I_{shift} \qquad (6)$$

where I_{shift} is a shift index, and where I_{max} varies according to the spectral power of the audio signal. An important observation here is that a different set of code frequency indices I_1 and I_0 from input block to input block is selected for spectral modulation depending on the frequency index I_{max} of the corresponding input block. In this case, a code bit is coded as a single bit: however, the frequencies that are used to encode each bit hop from block to block.

Unlike many traditional coding methods, such as Frequency Shift Keying (FSK) or Phase Shift Keying (PSK), the present invention does not rely on a single fixed frequency. Accordingly, a "frequency-hopping" effect is created similar to that seen in spread spectrum modulation systems. However, unlike spread spectrum, the object of varying the coding

frequencies of the present invention is to avoid the use of a constant code frequency which may render it audible.

For either of the two code frequencies selection approaches (a) and (b) described above, there are at least four modulation methods that can be implemented at a step 56 in order to encode a binary bit of data in an audio block, i.e., amplitude modulation, modulation by frequency swapping, phase modulation, and odd/even index modulation. These four methods of modulation are separately described below.

(i) Amplitude Modulation

In order to code a binary '1' using amplitude modulation, the spectral power at I_1 is increased to a level such that it constitutes a maximum in its corresponding neighborhood of frequencies. The neighborhood of indices corresponding to this neighborhood of frequencies is analyzed at a step 48 in order to determine how much the code frequencies f_1 and f_0 must be boosted and attenuated, respectively, so that they are detectable by the decoder 26. For index I_1, the neighborhood may preferably extend from $I_1 - 2$ to $I_1 + 2$, and is constrained to cover a narrow enough range of frequencies that the neighborhood of I_1 does not overlap the neighborhood of I_0. Simultaneously, the spectral power at I_0 is modified in order to make it a minimum in its neighborhood of indices ranging from $I_0 - 2$ to $I_0 + 2$. Conversely, in order to code a binary

'0' using amplitude modulation, the power at I_0 is boosted and the power at I_1 is attenuated in their corresponding neighborhoods.

As an example, Figure 3 shows a typical spectrum 50 of a N_C sample audio block plotted over a range of frequency index from forty five to seventy seven. A spectrum 52 shows the audio block after coding of a '1' bit, and a spectrum 54 shows the audio block before coding. In this particular instance of encoding a '1' bit according to code frequency selection approach (a), the hop sequence value is five which yields a mid-frequency index of fifty eight. The values for I_1 and I_0 are fifty three and sixty three, respectively. The spectral amplitude at fifty three is then modified at a step 56 of Figure 2 in order to make it a maximum within its neighborhood of indices. The amplitude at sixty three already constitutes a minimum and, therefore, only a small additional attenuation is applied at the step 56.

The spectral power modification process requires the computation of four values each in the neighborhood of I_1 and I_0. For the neighborhood of I_1 these four values are as follows: (1) I_{max1} which is the index of the frequency in the neighborhood of I_1 having maximum power; (2) P_{max1} which is the spectral power at I_{max1}; (3) I_{min1} which is the index of the frequency in the neighborhood of I_1 having minimum power; and

(4) P_{min1} which is the spectral power at I_{min1}. Corresponding values for the I_0 neighborhood are I_{max0}, P_{max0}, I_{min0}, and P_{min0}.

If $I_{max1} = I_1$, and if the binary value to be coded is a '1,' only a token increase in P_{max1} (i.e., the power at I_1) is required at the step 56. Similarly, if $I_{min0} = I_0$, then only a token decrease in P_{max0} (i.e., the power at I_0) is required at the step 56. When P_{max1} is boosted, it is multiplied by a factor 1 + A at the step 56, where *A* is in the range of about 1.5 to about 2.0. The choice of *A* is based on experimental audibility tests combined with compression survivability tests. The condition for imperceptibility requires a low value for *A*, whereas the condition for compression survivability requires a large value for A. A fixed value of A may not lend itself to only a token increase or decrease of power. Therefore, a more logical choice for *A* would be a value based on the local masking threshold. In this case, A is variable, and coding can be achieved with a minimal incremental power level change and yet survive compression.

In either case, the spectral power at I_1 is given by the following equation:

$$P_{I1} = (1 + A) \cdot P_{max1} \tag{7}$$

with suitable modification of the real and imaginary parts of the frequency component at I_1. The real and imaginary parts

are multiplied by the same factor in order to keep the phase angle constant. The power at I_0 is reduced to a value corresponding to $(1 + A)^{-1} P_{min0}$ in a similar fashion.

The Fourier Transform of the block to be coded as determined at the step 44 also contains negative frequency components with indices ranging in index values from -256 to -1. Spectral amplitudes at frequency indices $-I_1$ and $-I_0$ must be set to values representing the complex conjugate of amplitudes at I_1 and I_0, respectively, according to the following equations:

$$Re[f(-I_1)] = Re[f(I_1)] \tag{8}$$

$$Im[f(-I_1)] = -Im[f(I_1)] \tag{9}$$

$$Re[f(-I_0)] = Re[f(I_0)] \tag{10}$$

$$Im[f(-I_0)] = -Im[f(I_0)] \tag{11}$$

where f(I) is the complex spectral amplitude at index I.

Compression algorithms based on the effect of masking modify the amplitude of individual spectral components by means of a bit allocation algorithm. Frequency bands subjected to a high level of masking by the presence of high spectral energies in neighboring bands are assigned fewer

bits, with the result that their amplitudes are coarsely quantized. However, the decompressed audio under most conditions tends to maintain relative amplitude levels at frequencies within a neighborhood. The selected frequencies in the encoded audio stream which have been amplified or attenuated at the step 56 will, therefore, maintain their relative positions even after a compression/decompression process.

It may happen that the Fourier Transform $\Im\{v(t)\}$ of a block may not result in a frequency component of sufficient amplitude at the frequencies f_1 and f_0 to permit encoding of a bit by boosting the power at the appropriate frequency. In this event, it is preferable not to encode this block and to instead encode a subsequent block where the power of the signal at the frequencies f_1 and f_0 is appropriate for encoding.

(ii) Modulation by Frequency Swapping

In this approach, which is a variation of the amplitude modulation approach described above in section (i), the spectral amplitudes at I_1 and I_{max1} are swapped when encoding a one bit while retaining the original phase angles at I_1 and I_{max1}. A similar swap between the spectral amplitudes at I_0 and I_{max0} is also performed. When encoding a zero bit, the roles of I_1 and I_0 are reversed as in the case of amplitude modulation. As in the previous case, swapping is also applied to

the corresponding negative frequency indices. This encoding approach results in a lower audibility level because the encoded signal undergoes only a minor frequency distortion. Both the unencoded and encoded signals have identical energy values.

(iii) Phase Modulation

The phase angle associated with a spectral component I_0 is given by the following equation:

$$\phi_0 = \tan^{-1}\frac{Im[f(I_o)]}{Re[f(I_0)]} \tag{12}$$

where $0 \le \phi_0 \le 2\pi$. The phase angle associated with I_1 can be computed in a similar fashion. In order to encode a binary number, the phase angle of one of these components, usually the component with the lower spectral amplitude, can be modified to be either in phase (i.e., 0^0) or out of phase (i.e., 180^0) with respect to the other component, which becomes the reference. In this manner, a binary 0 may be encoded as an in-phase modification and a binary 1 encoded as an out-of-phase modification. Alternatively, a binary 1 may be encoded as an in-phase modification and a binary 0 encoded as an out-of-phase modification. The phase angle of the component that

is modified is designated ϕ_M, and the phase angle of the other component is designated ϕ_R. Choosing the lower amplitude component to be the modifiable spectral component minimizes the change in the original audio signal.

In order to accomplish this form of modulation, one of the spectral components may have to undergo a maximum phase change of 180^0, which could make the code audible. In practice, however, it is not essential to perform phase modulation to this extent, as it is only necessary to ensure that the two components are either "close" to one another in phase or "far" apart. Therefore, at the step 48, a phase neighborhood extending over a range of $\pm\pi/4$ around ϕ_R, the reference component, and another neighborhood extending over a range of $\pm\pi/4$ around $\phi_R + \pi$ may be chosen. The modifiable spectral component has its phase angle ϕ_M modified at the step 56 so as to fall into one of these phase neighborhoods depending upon whether a binary '0' or a binary '1' is being encoded. If a modifiable spectral component is already in the appropriate phase neighborhood, no phase modification may be necessary. In typical audio streams, approximately 30% of the segments are "self-coded" in this manner and no modulation is required. The inverse Fourier Transform is determined at the step 62.

(iv) Odd/Even Index Modulation

In this odd/even index modulation approach, a single code frequency index, I_1, selected as in the case of the other modulation schemes, is used. A neighborhood defined by indexes I_1, $I_1 + 1$, $I_1 + 2$, and $I_1 + 3$, is analyzed to determine whether the index I_M corresponding to the spectral component having the maximum power in this neighborhood is odd or even. If the bit to be encoded is a '1' and the index I_M is odd, then the block being coded is assumed to be "auto-coded." Otherwise, an odd-indexed frequency in the neighborhood is selected for amplification in order to make it a maximum. A bit '0' is coded in a similar manner using an even index. In the neighborhood consisting of four indexes, the probability that the parity of the index of the frequency with maximum spectral power will match that required for coding the appropriate bit value is 0.25. Therefore, 25% of the blocks, on an average, would be auto-coded. This type of coding will significantly decrease code audibility.

A practical problem associated with block coding by either amplitude or phase modulation of the type described above is that large discontinuities in the audio signal can arise at a boundary between successive blocks. These sharp transitions can render the code audible. In order to eliminate these sharp transitions, the time-domain signal v(t) can be multiplied by a smooth envelope or window function w(t) at

the step 42 prior to performing the Fourier Transform at the step 44. No window function is required for the modulation by frequency swapping approach described herein. The frequency distortion is usually small enough to produce only minor edge discontinuities in the time domain between adjacent blocks.

The window function w(t) is depicted in Figure 4. Therefore, the analysis performed at the step 54 is limited to the central section of the block resulting from $\Im_m\{v(t)w(t)\}$. The required spectral modulation is implemented at the step 56 on the transform $\Im\{v(t)w(t)\}$.

The modified frequency spectrum which now contains the binary code (either '0' or '1') is subjected to an inverse transform operation at a step 62 in order to obtain the encoded time domain signal, as will be discussed below. Following the step 62, the coded time domain signal is determined at a step 64 according to the following equation:

$$v_0(t) = v(t) + (\Im_m^{-1}(v(t)w(t)) - v(t)w(t)) \tag{13}$$

where the first part of the right hand side of equation (13) is the original audio signal v(t), where the second part of the right hand side of equation (13) is the encoding, and

where the left hand side of equation (13) is the resulting encoded audio signal $v_0(t)$.

While individual bits can be coded by the method described thus far, practical decoding of digital data also requires (i) synchronization, so as to locate the start of data, and (ii) built-in error correction, so as to provide for reliable data reception. The raw bit error rate resulting from coding by spectral modulation is high and can typically reach a value of 20%. In the presence of such error rates, both synchronization and error-correction may be achieved by using pseudo-noise (PN) sequences of ones and zeroes. A PN sequence can be generated, for example, by using an m-stage shift register 58 (where m is three in the case of Figure 5) and an exclusive-OR gate 60 as shown in Figure 5. For convenience, an n-bit PN sequence is referred to herein as a PNn sequence. For an N_{PN} bit PN sequence, an m-stage shift register is required operating according to the following equation:

$$N_{PN} = 2^m - 1 \tag{14}$$

where m is an integer. With m = 3, for example, the 7-bit PN sequence (PN7) is 1110100. The particular sequence depends upon an initial setting of the shift register 58. In one robust version of the encoder 12, each individual bit of data is represented by this PN sequence - i.e., 1110100 is used

for a bit '1,' and the complement 0001011 is used for a bit '0.' The use of seven bits to code each bit of code results in extremely high coding overheads.

An alternative method uses a plurality of PN15 sequences, each of which includes five bits of code data and 10 appended error correction bits. This representation provides a Hamming distance of 7 between any two 5-bit code data words. Up to three errors in a fifteen bit sequence can be detected and corrected. This PN15 sequence is ideally suited for a channel with a raw bit error rate of 20%.

In terms of synchronization, a unique synchronization sequence 66 (Figure 7a) is required for synchronization in order to distinguish PN15 code bit sequences 74 from other bit sequences in the coded data stream. In a preferred embodiment shown in Figure 7b, the first code block of the synchronization sequence 66 uses a "triple tone" 70 of the synchronization sequence in which three frequencies with indices I_0, I_1, and I_{mid} are all amplified sufficiently that each becomes a maximum in its respective neighborhood, as depicted by way of example in Figure 6. It will be noted that, although it is preferred to generate the triple tone 70 by amplifying the signals at the three selected frequencies to be relative maxima in their respective frequency neighborhoods, those signals could instead be locally attenuated so that the three associated local extreme values comprise three

local minima. It should be noted that any combination of local maxima and local minima could be used for the triple tone 70. However, because program audio signals include substantial periods of silence, the preferred approach involves local amplification rather than local attenuation. Being the first bit in a sequence, the hop sequence value for the block from which the triple tone 70 is derived is two and the mid-frequency index is fifty-five. In order to make the triple tone block truly unique, a shift index of seven may be chosen instead of the usual five. The three indices I_0, I_1, and I_{mid} whose amplitudes are all amplified are forty-eight, sixty-two and fifty-five as shown in Figure 6. (In this example, $I_{mid} = H_S + 53 = 2 + 53 = 55$.) The triple tone 70 is the first block of the fifteen block sequence 66 and essentially represents one bit of synchronization data. The remaining fourteen blocks of the synchronization sequence 66 are made up of two PN7 sequences: 1110100, 0001011. This makes the fifteen synchronization blocks distinct from all the PN sequences representing code data.

As stated earlier, the code data to be transmitted is converted into five bit groups, each of which is represented by a PN15 sequence. As shown in Figure 7a, an unencoded block 72 is inserted between each successive pair of PN sequences 74. During decoding, this unencoded block 72 (or gap) between neighboring PN sequences 74 allows precise syn-

chronizing by permitting a search for a correlation maximum across a range of audio samples.

In the case of stereo signals, the left and right channels are encoded with identical digital data. In the case of mono signals, the left and right channels are combined to produce a single audio signal stream. Because the frequencies selected for modulation are identical in both channels, the resulting monophonic sound is also expected to have the desired spectral characteristics so that, when decoded, the same digital code is recovered.

DECODING THE SPECTRALLY MODULATED SIGNAL

In most instances, the embedded digital code can be recovered from the audio signal available at the audio output 28 of the receiver 20. Alternatively, or where the receiver 20 does not have an audio output 28, an analog signal can be reproduced by means of the microphone 30 placed in the vicinity of the speakers 24. In the case where the microphone 30 is used, or in the case where the signal on the audio output 28 is analog, the decoder 20 converts the analog audio to a sampled digital output stream at a preferred sampling rate matching the sampling rate of the encoder 12. In decoding systems where there are limitations in terms of memory and computing power, a half-rate sampling could be used. In the case of half-rate sampling, each code block would consist of

$N_c/2$ = 256 samples, and the resolution in the frequency domain (i.e., the frequency difference between successive spectral components) would remain the same as in the full sampling rate case. In the case where the receiver 20 provides digital outputs, the digital outputs are processed directly by the decoder 26 without sampling but at a data rate suitable for the decoder 26.

The task of decoding is primarily one of matching the decoded data bits with those of a PN15 sequence which could be either a synchronization sequence or a code data sequence representing one or more code data bits. The case of amplitude modulated audio blocks is considered here. However, decoding of phase modulated blocks is virtually identical, except for the spectral analysis, which would compare phase angles rather than amplitude distributions, and decoding of index modulated blocks would similarly analyze the parity of the frequency index with maximum power in the specified neighborhood. Audio blocks encoded by frequency swapping can also be decoded by the same process.

In a practical implementation of audio decoding, such as may be used in a home audience metering system, the ability to decode an audio stream in real-time is highly desirable. It is also highly desirable to transmit the decoded data to a central office. The decoder 26 may be arranged to run the decoding algorithm described below on Digi-

tal Signal Processing (DSP) based hardware typically used in such applications. As disclosed above, the incoming encoded audio signal may be made available to the decoder 26 from either the audio output 28 or from the microphone 30 placed in the vicinity of the speakers 24. In order to increase processing speed and reduce memory requirements, the decoder 26 may sample the incoming encoded audio signal at half (24 kHz) of the normal 48 kHz sampling rate.

Before recovering the actual data bits representing code information, it is necessary to locate the synchronization sequence. In order to search for the synchronization sequence within an incoming audio stream, blocks of 256 samples, each consisting of the most recently received sample and the 255 prior samples, could be analyzed. For real-time operation, this analysis, which includes computing the Fast Fourier Transform of the 256 sample block, has to be completed before the arrival of the next sample. Performing a 256-point Fast Fourier Transform on a 40 MHZ DSP processor takes about 600 microseconds. However, the time between samples is only 40 microseconds, making real time processing of the incoming coded audio signal as described above impractical with current hardware.

Therefore, instead of computing a normal Fast Fourier Transform on each 256 sample block, the decoder 26 may be arranged to achieve real-time decoding by implementing an

incremental or sliding Fast Fourier Transform routine 100 (Figure 8) coupled with the use of a status information array SIS that is continuously updated as processing progresses. This array comprises p elements SIS[0] to SIS[p-1]. If p = 64, for example, the elements in the status information array SIS are SIS[0] to SIS[63].

Moreover, unlike a conventional transform which computes the complete spectrum consisting of 256 frequency "bins," the decoder 26 computes the spectral amplitude only at frequency indexes that belong to the neighborhoods of interest, i.e., the neighborhoods used by the encoder 12. In a typical example, frequency indexes ranging from 45 to 70 are adequate so that the corresponding frequency spectrum contains only twenty-six frequency bins. Any code that is recovered appears in one or more elements of the status information array SIS as soon as the end of a message block is encountered.

Additionally, it is noted that the frequency spectrum as analyzed by a Fast Fourier Transform typically changes very little over a small number of samples of an audio stream. Therefore, instead of processing each block of 256 samples consisting of one "new" sample and 255 "old" samples, 256 sample blocks may be processed such that, in each block of 256 samples to be processed, the last k samples are "new" and the remaining 256-k samples are from a previous analysis. In the

case where $k = 4$, processing speed may be increased by skipping through the audio stream in four sample increments, where a skip factor k is defined as $k = 4$ to account for this operation.

Each element SIS[p] of the status information array SIS consists of five members: a previous condition status PCS, a next jump index JI, a group counter GC, a raw data array DA, and an output data array OP. The raw data array DA has the capacity to hold fifteen integers. The output data array OP stores ten integers, with each integer of the output data array OP corresponding to a five bit number extracted from a recovered PN15 sequence. This PN15 sequence, accordingly, has five actual data bits and ten other bits. These other bits may be used, for example, for error correction. It is assumed here that the useful data in a message block consists of 50 bits divided into 10 groups with each group containing 5 bits, although a message block of any size may be used.

The operation of the status information array SIS is best explained in connection with Figure 8. An initial block of 256 samples of received audio is read into a buffer at a processing stage 102. The initial block of 256 samples is analyzed at a processing stage 104 by a conventional Fast Fourier Transform to obtain its spectral power distribution. All subsequent transforms implemented by the routine 100 use

the high-speed incremental approach referred to above and described below.

In order to first locate the synchronization sequence, the Fast Fourier Transform corresponding to the initial 256 sample block read at the processing stage 102 is tested at a processing stage 106 for a triple tone, which represents the first bit in the synchronization sequence. The presence of a triple tone may be determined by examining the initial 256 sample block for the indices I_0, I_1, and I_{mid} used by the encoder 12 in generating the triple tone, as described above. The SIS[*p*] element of the SIS array that is associated with this initial block of 256 samples is SIS[0], where the status array index *p* is equal to 0. If a triple tone is found at the processing stage 106, the values of certain members of the SIS[0] element of the status information array SIS are changed at a processing stage 108 as follows: the previous condition status PCS, which is initially set to 0, is changed to a 1 indicating that a triple tone was found in the sample block corresponding to SIS[0]; the value of the next jump index JI is incremented to 1; and, the first integer of the raw data member DA[0] in the raw data array DA is set to the value (0 or 1) of the triple tone. In this case, the first integer of the raw data member DA[0] in the raw data array DA is set to 1 because it is assumed in this analysis that the triple tone is the equivalent of a 1 bit. Also, the status

array index p is incremented by one for the next sample block. If there is no triple tone, none of these changes in the SIS[0] element are made at the processing stage 108, but the status array index p is still incremented by one for the next sample block. Whether or not a triple tone is detected in this 256 sample block, the routine 100 enters an incremental FFT mode at a processing stage 110.

Accordingly, a new 256 sample block increment is read into the buffer at a processing stage 112 by adding four new samples to, and discarding the four oldest samples from, the initial 256 sample block processed at the processing stages 102 - 106. This new 256 sample block increment is analyzed at a processing stage 114 according to the following steps:

<u>STEP 1</u>: the skip factor k of the Fourier Transform is applied according to the following equation in order to modify each frequency component $F_{old}(u_0)$ of the spectrum corresponding to the initial sample block in order to derive a corresponding intermediate frequency component $F_1(u_0)$:

$$F_1(u_0) = F_{old}(u_0)\exp-(\frac{2\pi u_0 k}{256}) \qquad (15)$$

where u_0 is the frequency index of interest. In accordance with the typical example described above, the frequency index u_0 varies from 45 to 70. It should be noted that this first step involves multiplication of two complex numbers.

STEP 2: the effect of the first four samples of the old 256 sample block is then eliminated from each $F_1(u_0)$ of the spectrum corresponding to the initial sample block and the effect of the four new samples is included in each $F_1(u_0)$ of the spectrum corresponding to the current sample block increment in order to obtain the new spectral amplitude $F_{new}(u_o)$ for each frequency index u_0 according to the following equation:

$$F_{new}(u_0) = F_1(u_0) + \sum_{m=1}^{m=4} (f_{new}(m) - f_{old}(m))\exp-(\frac{2\pi u_0(k-m+1)}{256}) \quad (16)$$

where f_{old} and f_{new} are the time-domain sample values. It should be noted that this second step involves the addition of a complex number to the summation of a product of a real number and a complex number. This computation is repeated across the frequency index range of interest (for example, 45 to 70).

STEP 3: the effect of the multiplication of the 256 sample block by the window function in the encoder 12 is then taken into account. That is, the results of step 2 above are not confined by the window function that is used in the encoder 12. Therefore, the results of step 2 preferably should be multiplied by this window function. Because multiplication in the time domain is equivalent to a convolution of the spectrum by the Fourier Transform of the window function, the results from the second step may be convolved with the window function. In this case, the preferred window function for this operation is the following well known "raised cosine" function which has a narrow 3-index spectrum with amplitudes (-0.50, 1, +0.50):

$$w(t) = \frac{1}{2}[1 - \cos(\frac{2\pi t}{T_W})] \tag{17}$$

where T_W is the width of the window in the time domain. This "raised cosine" function requires only three multiplication and addition operations involving the real and imaginary parts of the spectral amplitude. This operation significantly improves computational speed. This step is not required for the case of modulation by frequency swapping.

STEP 4: the spectrum resulting from step 3 is then examined for the presence of a triple tone. If a triple tone is found, the values of certain members of the SIS[1] element of the status information array SIS are set at a processing stage 116 as follows: the previous condition status PCS, which is initially set to 0, is changed to a 1; the value of the next jump index JI is incremented to 1; and, the first integer of the raw data member DA[1] in the raw data array DA is set to 1. Also, the status array index *p* is incremented by one. If there is no triple tone, none of these changes are made to the members of the structure of the SIS[1] element at the processing stage 116, but the status array index *p* is still incremented by one.

Because *p* is not yet equal to 64 as determined at a processing stage 118 and the group counter GC has not accumulated a count of 10 as determined at a processing stage 120, this analysis corresponding to the processing stages 112 - 120 proceeds in the manner described above in four sample increments where p is incremented for each sample increment. When SIS[63] is reached where p = 64, p is reset to 0 at the processing stage 118 and the 256 sample block increment now in the buffer is exactly 256 samples away from the location in the audio stream at which the SIS[0] element was last updated. Each time p reaches 64, the SIS array represented by the SIS[0] - SIS[63] elements is examined to determine whether the

previous condition status PCS of any of these elements is one indicating a triple tone. If the previous condition status PCS of any of these elements corresponding to the current 64 sample block increments is not one, the processing stages 112 - 120 are repeated for the next 64 block increments. (Each block increment comprises 256 samples.)

Once the previous condition status PCS is equal to 1 for any of the SIS[0] - SIS[63] elements corresponding to any set of 64 sample block increments, and the corresponding raw data member DA[*p*] is set to the value of the triple tone bit, the next 64 block increments are analyzed at the processing stages 112 - 120 for the next bit in the synchronization sequence.

Each of the new block increments beginning where p was reset to 0 is analyzed for the next bit in the synchronization sequence. This analysis uses the second member of the hop sequence H_S because the next jump index JI is equal to 1. From this hop sequence number and the shift index used in encoding, the I_1 and I_0 indexes can be determined, for example from equations (2) and (3). Then, the neighborhoods of the I_1 and I_0 indexes are analyzed to locate maximums and minimums in the case of amplitude modulation. If, for example, a power maximum at I_1 and a power minimum at I_0 are detected, the next bit in the synchronization sequence is taken to be 1. In order to allow for some variations in the signal that may

arise due to compression or other forms of distortion, the index for either the maximum power or minimum power in a neighborhood is allowed to deviate by 1 from its expected value. For example, if a power maximum is found in the index I_1, and if the power minimum in the index I_0 neighborhood is found at $I_0 - 1$, instead of I_0, the next bit in the synchronization sequence is still taken to be 1. On the other hand, if a power minimum at I_1 and a power maximum at I_0 are detected using the same allowable variations discussed above, the next bit in the synchronization sequence is taken to be 0. However, if none of these conditions are satisfied, the output code is set to -1, indicating a sample block that cannot be decoded. Assuming that a 0 bit or a 1 bit is found, the second integer of the raw data member DA[1] in the raw data array DA is set to the appropriate value, and the next jump index JI of SIS[0] is incremented to 2, which corresponds to the third member of the hop sequence H_S. From this hop sequence number and the shift index used in encoding, the I_1 and I_0 indexes can be determined. Then, the neighborhoods of the I_1 and I_0 indexes are analyzed to locate maximums and minimums in the case of amplitude modulation so that the value of the next bit can be decoded from the third set of 64 block increments, and so on for fifteen such bits of the synchronization sequence. The fifteen bits stored in the raw data array DA may then be compared with a reference synchronization sequence

to determine synchronization. If the number of errors between the fifteen bits stored in the raw data array DA and the reference synchronization sequence exceeds a previously set threshold, the extracted sequence is not acceptable as a synchronization, and the search for the synchronization sequence begins anew with a search for a triple tone.

If a valid synchronization sequence is thus detected, there is a valid synchronization, and the PN15 data sequences may then be extracted using the same analysis as is used for the synchronization sequence, except that detection of each PN15 data sequence is not conditioned upon detection of the triple tone which is reserved for the synchronization sequence. As each bit of a PN15 data sequence is found, it is inserted as a corresponding integer of the raw data array DA. When all integers of the raw data array DA are filled, (i) these integers are compared to each of the thirty-two possible PN15 sequences, (ii) the best matching sequence indicates which 5-bit number to select for writing into the appropriate array location of the output data array OP, and (iii) the group counter GC member is incremented to indicate that the first PN15 data sequence has been successfully extracted. If the group counter GC has not yet been incremented to 10 as determined at the processing stage 120, program flow returns to the processing stage 112 in order to decode the next PN15 data sequence.

When the group counter GC has incremented to 10 as determined at the processing stage 120, the output data array OP, which contains a full 50-bit message, is read at a processing stage 122. The total number of samples in a message block is 45,056 at a half-rate sampling frequency of 24 kHz. It is possible that several adjacent elements of the status information array SIS, each representing a message block separated by four samples from its neighbor, may lead to the recovery of the same message because synchronization may occur at several locations in the audio stream which are close to one another. If all these messages are identical, there is a high probability that an error-free code has been received.

Once a message has been recovered and the message has been read at the processing stage 122, the previous condition status PCS of the corresponding SIS element is set to 0 at a processing stage 124 so that searching is resumed at a processing stage 126 for the triple tone of the synchronization sequence of the next message block.

MULTI-LEVEL CODING

Often there is a need to insert more than one code message into the same audio stream. For example in a television program distribution environment, the network originator of the program may insert its identification code and time stamp, and a network affiliated station carrying this program

may also insert its own identification code. In addition, an advertiser or sponsor may wish to have its code added. It is noted that the network originator, the network affiliated station, and the advertiser are at different distribution levels between audio origination and audio reception by the consumer. There are a number of methods of accommodating multi-level encoding in order to designate more than one distributor of the audio.

(i) Bit Reservation

In order to accommodate multi-level coding, 48 bits in a 50-bit system can be used for the code and the remaining 2 bits can be used for level specification. Usually the first program material generator, say the network, will insert codes in the audio stream. Its first message block would have the level bits set to 00, and only a synchronization sequence and the 2 level bits are set for the second and third message blocks in the case of a three level system. For example, the level bits for the second and third messages may be both set to 11 indicating that the actual data areas have been left unused.

The network affiliated station can now enter its code with a decoder/encoder combination that would locate the synchronization of the second message block with the 11 level setting. This station inserts its code in the data area of

this block and sets the level bits to 01. The next level encoder inserts its code in the third message block's data area and sets the level bits to 10. During decoding, the level bits distinguish each message level category.

(ii) Frequency Multiplexing

In frequency multiplexing, each code level (e.g., network, affiliate, advertiser) is assigned to a different frequency band in the spectrum. In determining the size of a frequency band and, therefore, the number of bands that may be coded, it is noted that each code level generally requires a minimum of eighteen consecutive spectral lines when using the coding methods described herein. This requirement follows from the way in which a triple tone is coded. That is, in coding a triple tone, the frequencies corresponding to indices I_1, I_0, and I_{mid} are all amplified. Because I_1 = forty-eight and I_0 = sixty-two, the two outer frequencies corresponding to I_1 and I_0 are separated by fourteen spectral lines. In addition, the neighborhoods defined for these frequencies extend two spectral lines on either side of these two frequencies for a total of eighteen spectral lines.

At a sampling rate of 48 kHz and 512 samples per block, eighteen spectral lines correspond to a spectral width of 1.69 kHz. In order to insert a code, there must be enough energy within this 1.69 kHz band to provide masking for the

code signal. Three levels of code can be inserted in an audio signal typically having a bandwidth of 8 kHz by choosing the following bands: 2.9 kHz to 4.6 kHz for a first level of coding; 4.6 kHz to 6.3 kHz for a second level of coding; and, 6.3 kHz to 8.0 kHz for a third level of coding. However, it should be noted that audio consisting of speech usually has a bandwidth lower than 5 kHz and may, therefore, support only a single level of code.

(iii) Primary/Secondary Encoding

In this method of encoding, two types of encoders, a primary encoder and one or more secondary encoders, may be used to insert different levels of code. The various levels of code can be arranged hierarchically in such a manner that the primary encoder inserts at least the synchronization sequence and may also insert one of the levels, such as the highest level, of code. During encoding, and preferably prior to insertion of the synchronization sequence, the primary encoder leaves a predetermined number of audio blocks uncoded to permit the secondary encoders to insert their assigned levels of code. Accordingly, the secondary encoders have the capability to both decode and encode audio such that they first locate the synchronization sequence inserted by the primary encoder, and then determine their assigned positions in the audio stream for insertion of their corresponding

codes. In the decoding process, the synchronization sequence is first detected, and then the several levels of codes are recovered sequentially.

CODE ERASURE AND OVERWRITE

It may also be necessary to provide a means of erasing a code or to erase and overwrite a code. Erasure may be accomplished by detecting the triple tone/synchronization sequence using a decoder and by then modifying at least one of the triple tone frequencies such that the code is no longer recoverable. Overwriting involves extracting the synchronization sequence in the audio, testing the data bits in the data area and inserting a new bit only in those blocks that do not have the desired bit value. The new bit is inserted by amplifying and attenuating appropriate frequencies in the data area.

DELAY COMPENSATION

In a practical implementation of the encoder 12, N_C samples of audio, where N_C is typically 512, are processed at any given time. In order to achieve operation with a minimum amount of throughput delay, the following four buffers are used: input buffers IN0 and IN1, and output buffers OUT0 and OUT1. Each of these buffers can hold N_C samples. While samples in the input buffer IN0 are being processed, the input

buffer IN1 receives new incoming samples. The processed output samples from the input buffer IN0 are written into the output buffer OUT0, and samples previously encoded are written to the output from the output buffer OUT1. When the operation associated with each of these buffers is completed, processing begins on the samples stored in the input buffer IN1 while the input buffer IN0 starts receiving new data. Data from the output buffer OUT0 are now written to the output. This cycle of switching between the pair of buffers in the input and output sections of the encoder continues as long as new audio samples arrive for encoding. It is clear that a sample arriving at the input suffers a delay equivalent to the time duration required to fill two buffers at the sampling rate of 48 kHz before its encoded version appears at the output. This delay is approximately 22 ms. When the encoder 12 is used in a television system environment, it is necessary to compensate for this delay in order to maintain synchronization between video and audio.

Such a compensation arrangement is shown in Figure 9. As shown in Figure 9, an encoding arrangement 200, which may be used for the elements 12, 14, and 18 in Figure 1, is arranged to receive either analog video and audio inputs or digital video and audio inputs. Analog video and audio inputs are supplied to corresponding video and audio analog to digital converters 202 and 204. The audio samples from the audio

analog to digital converter 204 are provided to an audio encoder 206 which may be of known design or which may be arranged as disclosed above. The digital audio input is supplied directly to the audio encoder 206. Alternatively, if the input digital bit stream is a combination of digital video and audio bit stream portions, the input digital bit stream is provided to a demultiplexer 208 which separates the digital video and audio portions of the input digital bit stream and supplies the separated digital audio portion to the audio encoder 206.

Because the audio encoder 206 imposes a delay on the digital audio bit stream as discussed above relative to the digital video bit stream, a delay 210 is introduced in the digital video bit stream. The delay imposed on the digital video bit stream by the delay 210 is equal to the delay imposed on the digital audio bit stream by the audio encoder 206. Accordingly, the digital video and audio bit streams downstream of the encoding arrangement 200 will be synchronized.

In the case where analog video and audio inputs are provided to the encoding arrangement 200, the output of the delay 210 is provided to a video digital to analog converter 212 and the output of the audio encoder 206 is provided to an audio digital to analog converter 214. In the case where separate digital video and audio bit streams are provided to

the encoding arrangement 200, the output of the delay 210 is provided directly as a digital video output of the encoding arrangement 200 and the output of the audio encoder 206 is provided directly as a digital audio output of the encoding arrangement 200. However, in the case where a combined digital video and audio bit stream is provided to the encoding arrangement 200, the outputs of the delay 210 and of the audio encoder 206 are provided to a multiplexer 216 which recombines the digital video and audio bit streams as an output of the encoding arrangement 200.

As explained above, there may be some instances where the arrangement described above can result in undesirable audibility of the ancillary code inserted into a program audio signal. Two such instances and exemplary solutions to these two instances are described below.

Controlling Code Audibility Using an Audio Quality Measure (AQM)

One example of audio material that is difficult to inaudibly encode is instrumental music characterized by strong harmonics or by a strong fundamental frequency in the code frequency band. Shifting the frequency maxima and minima in such cases can lead to audible distortion. Therefore, an audibility score, which is designated herein as the audio quality measure (AQM), can be computed in order to determine when instances of potentially audible code segments occur.

AQM computation may be based on psycho-acoustic models that are widely used in audio compression algorithms such as Dolby's AC-3, MPEG-2 Layers I, II, or III, or MPEG-AAC. The AQM computation discussed below is based on MPEG-AAC. However, the AQM computation may be based any of these audio compression algorithms. (For example, in the Dolby AC-3 audio compression method, a Modified Discrete Cosine Transform (MDCT) spectrum is used for computing the masking levels.)

Let it be assumed that blocks of 512 samples at a 48 kHz sampling rate are used to compute the AQM. The frequency space extending from 0 to 24 kHz is divided into 42 critical bands. Prior to encoding a block of audio as described above, the spectral energy $E_0[b]$ in each critical band, where b is the band index, is computed by the encoder 12 at the step 48 in accordance with the following equation:

$$E_0[b] = \sum_{f=fi}^{f=fl} A^2[f] \tag{18}$$

where A[f] is the amplitude at a frequency component f in the corresponding critical band of the audio block, fi is the initial frequency component in the corresponding critical band of the audio block, and fl is the last frequency component in the corresponding critical band of the audio block.

A masking energy level $E_{MASK}[b]$ is also computed at the step 48 following the methodology described in ISO/IEC 13818-7:1997. The masking energy level $E_{MASK}[b]$ is the minimum change in energy within the band b that will be perceptible to the human ear.

If this block were to be coded by the spectral modulation procedure described earlier in this application, a new energy level value $E_C[b]$ for each band in the coded block will result and can be computed at the step 48 using equation (18).

The encoder 12 at the step 56 determines whether the change in energy of a band b given by $|E_C[b] - E_0[b]|$ is less than the masking energy level $E_{MASK}[b]$. If $|E_C[b] - E_0[b]|$ is less than $E_{MASK}[b]$, it can be assumed that there is adequate masking energy available in the band b to make the change resulting from coding imperceptible. Therefore, an aqm[b] for this band b is assumed to be zero. However, if $|E_C[b] - E_0[b]| \geq E_{MASK}[b]$ for the band b, the aqm for the band can be computed at the step 56 as follows:

$$aqm[b] = \frac{|E_C[b]-E_0[b]|}{E_{MASK}[b]} \tag{19}$$

The total AQM score for the whole block can be obtained at the step 56 from equation (19) by summing across all 42 critical bands according to the following equation:

$$AQM_{TOTAL} = \sum_{b=0}^{b=41} aqm[b] \tag{20}$$

If it is determined at the step 56 that AQM_{TOTAL} is greater than a predetermined threshold AQM_{THRESH}, then the corresponding block is not considered to be suitable for encoding.

In practice, however, coding of a single audio block, or even several audio blocks, whose $AQM_{TOTAL} > AQM_{THRESH}$ and whose durations are each approximately 10 ms, may not result in an audible code. But if one such audio block occurs, it is likely to occur near in time to other such audio blocks with the result that, if a sufficient number of such audio blocks are grouped consecutively in a sequence, coding of one or more audio blocks in the sequence may well produce an audible code thereby degrading the quality of the original audio.

Therefore, in order to determine when to encode and when to suspend encoding, the encoder 12 at the step 56 maintains a count of audible blocks. If x out of y consecutive blocks prior to the current block fall in the audible code

category, then the encoder 12 at the step 56 suspends coding for all subsequent blocks of the current ancillary code message. If x is equal to 9 and y is equal to 16, for example, and if 9 out 16 such audio blocks are coded in spite of the audibility scores being high, an audible code is likely to result. Therefore, in order to successfully encode a 50 bit ancillary code message, a sequence of z audio blocks is required, where the sequence of z audio blocks has less than x audible blocks in any consecutive y block segment.

In addition, encoding of any individual audio block may be inhibited if the AQM score for this individual audio block exceeds a threshold $AQM_{THRESH+}$ which is set higher than AQM_{THRESH}. Even though a single bit of code may be accordingly lost in such a case, the error correction discussed above will make it possible to still recover the ancillary code message.

Pre-echo Cancellation

Pre-echo is a well known phenomenon that is encountered in most or all block based audio processing operations such as compression. It also occurs in the case of audio encoding as described above. Pre-echo arises when the audio energy within a block is not uniformly distributed, but is instead concentrated in the latter half of the block. Pre-echo effects are most apparent in the extreme case when the first half of the audio block has a very low level of audio

and the second half of the audio block has a very high level of audio. As a result, a code signal, which is uniformly distributed across the entire audio block, has no masking energy available to make it inaudible during the first half of the audio block.

Therefore, each audio block, prior to coding at the step 56, is examined by the encoder 12 for the block's energy distribution characteristic. The energy in an audio block is computed by summing the squares of the amplitudes of the time domain samples. Then, if the ratio of the energy E_1 in a first part of the audio block to the energy E_2 in the remaining part of the audio block is below a threshold, a code is not inserted in the audio block. The energy E_1 and the energy E_2 are calculated according to the following equations:

$$E_1 = \sum_{s=o}^{s=d} A^2[s] \tag{21}$$

and

$$E_2 = \sum_{s=d+1}^{s=S} A^2[s] \tag{22}$$

where A[s] is the amplitude of a sample s, S is the total number of samples in a corresponding block of audio, and d divides the corresponding block of audio between samples in the first part of the block of audio and samples in the remaining part of the block of audio. For example, d may divide the block of audio between samples in the first quarter of the block of audio and samples in the last three quarters of the block of audio.

Certain modifications of the present invention have been discussed above. Other modifications will occur to those practicing in the art of the present invention. For example, according to the description above, the encoding arrangement 200 includes a delay 210 which imposes a delay on the video bit stream in order to compensate for the delay imposed on the audio bit stream by the audio encoder 206. However, some embodiments of the encoding arrangement 200 may include a video encoder 218, which may be of known design, in order to encode the video output of the video analog to digital converter 202, or the input digital video bit stream, or the output of the demultiplexer 208, as the case may be. When the video encoder 218 is used, the audio encoder 206 and/or the video encoder 218 may be adjusted so that the relative delay imposed on the audio and video bit streams is zero and so that the audio and video bit streams are thereby synchronized. In this case, the delay 210 is not necessary. Alternatively, the

delay 210 may be used to provide a suitable delay and may be inserted in either the video or audio processing so that the relative delay imposed on the audio and video bit streams is zero and so that the audio and video bit streams are thereby synchronized.

In still other embodiments of the encoding arrangement 200, the video encoder 218 and not the audio encoder 206 may be used. In this case, the delay 210 may be required in order to impose a delay on the audio bit stream so that the relative delay between the audio and video bit streams is zero and so that the audio and video bit streams are thereby synchronized.

Accordingly, the description of the present invention is to be construed as illustrative only and is for the purpose of teaching those skilled in the art the best mode of carrying out the invention. The details may be varied substantially without departing from the spirit of the invention, and the exclusive use of all modifications which are within the scope of the appended claims is reserved.

WHAT IS CLAIMED IS:

1. A method for encoding first and second blocks of audio with corresponding first and second binary code bits comprising the following steps:

a) selecting first and second frequencies from a frequency spectrum of the first block of audio;

b) modulating the audio based upon the first and second frequencies to thereby encode the first block of audio with the first binary code bit;

c) selecting third and fourth frequencies from a frequency spectrum of the second block of audio, wherein the third and fourth frequencies bear a predetermined offset relationship to the first and second frequencies; and,

d) modulating the audio based upon the third and fourth frequencies to thereby encode the second block of audio with the second binary code bit.

2. The method of claim 1 wherein:

step b) comprises the steps of (i) increasing the spectral power at one of the first and second frequencies so as to render the spectral power at the one of the first and second frequencies a local maximum, and (ii) decreasing the spectral power at the other of the first and second frequen-

cies so as to render the spectral power at the other of the first and second frequencies a local minimum; and,

step d) comprises the steps of (i) increasing the spectral power at one of the third and fourth frequencies so as to render the spectral power at the one of the third and fourth frequencies a local maximum, and (ii) decreasing the spectral power at the other of the third and fourth frequencies so as to render the spectral power at the other of the third and fourth frequencies a local minimum.

3. The method of claim 1 wherein:

step b) comprises the step of selectively changing a phase relationship between the first and second frequencies; and,

step d) comprises the step of selectively changing a phase relationship between the third and fourth frequencies.

4. The method of claim 1 wherein:

step b) comprises the step of swapping a spectral amplitude of at least one of the first and second frequencies with a spectral amplitude of a frequency having a maximum amplitude in a frequency neighborhood of the least one of the first and second frequencies; and,

step d) comprises the step of swapping a spectral amplitude of at least one of the third and fourth frequencies

with a spectral amplitude of a frequency having a maximum amplitude in a frequency neighborhood of the at least one of the third and fourth frequencies.

5. The method of claim 1 wherein the predetermined offset relationship is determined by a frequency hop sequence.

6. The method of claim 1 wherein the predetermined offset relationship is determined by hopping based on a low frequency maximum.

7. The method of claim 1 wherein the first and second frequencies are selected according to a reference frequency, a frequency hop sequence, and a predetermined shift index, and wherein the third and fourth frequencies are selected according to the reference frequency, the frequency hop sequence, and the predetermined shift index.

8. The method of claim 7 wherein the first and second frequencies are selected according to the following equations:

$$I_1 = I_{5k} + H_S - I_{shift}$$

and

$$I_0 = I_{5k} + H_S + I_{shift}$$

wherein I_1 and I_0 are indexes corresponding to the first and second frequencies respectively, wherein I_{5k} is an index corresponding to the reference frequency, wherein H_S is a component of the frequency hop sequence, and wherein I_{shift} is the predetermined shift index.

9. The method of claim 1 wherein the first and second frequencies are selected according to a reference frequency, a first low frequency maximum, and a predetermined shift index, and wherein the third and fourth frequencies are selected according to the reference frequency, a second low frequency maximum, and the predetermined shift index.

10. The method of claim 9 wherein the first and second frequencies are selected according to the following equations:

$$I_1 = I_{5k} + I_{\max} - I_{shift}$$

and

$$I_0 = I_{5k} + I_{\max} + I_{shift}$$

wherein I_1 and I_0 are indexes corresponding to the first and second frequencies respectively, wherein I_{5k} is an index corresponding to the reference frequency, wherein I_{max} is an index corresponding to a frequency at which the spectral power of the first block of audio is a maximum in a low frequency band of the first block of audio, and wherein I_{shift} is the predetermined shift index.

11. The method of claim 1 wherein a synchronization block is added to the signal, and wherein the synchronization block is characterized by a triple tone portion.

12. The method of claim 1 further comprising the following steps:

e) determining an audio quality measure AQM for the first block of audio;

f) comparing AQM to AQM_{THRESH}, wherein AQM_{THRESH} is a predetermined threshold audio quality measure reference; and,

g) if $AQM > AQM_{THRESH}$, inhibiting step b).

13. The method of claim 12 wherein the first block of audio has a spectral energy, wherein

$$aqm[b] = \frac{|E_C[b] - E_0[b]|}{E_{MASK}[b]}$$

wherein the first block of audio is divided into B frequency bands, wherein b is a variable representing a frequency band within the B frequency bands, wherein $E_0[b]$ is the spectral energy in the first block of audio without coding, wherein $E_C[b]$ is the spectral energy in the first block of audio with coding, and wherein

$$AQM = \sum_{b=0}^{b=B} aqm[b] .$$

14. The method of claim 1 further comprising the following steps:

e) determining an audio quality measure AQM for each of a plurality of blocks of audio;

f) comparing the AQM corresponding to each of the plurality of blocks of audio to AQM_{THRESH}, wherein AQM_{THRESH} is a predetermined threshold audio quality measure reference; and,

g) if $AQM > AQM_{THRESH}$ for x blocks of audio out of y blocks of audio of the plurality of blocks of audio, encoding of the audio with binary bits is suspended, wherein x and y are corresponding predetermined numbers of blocks of audio.

15. The method of claim 14 wherein y is a predetermined number of consecutive blocks of audio.

16. The method of claim 14 wherein encoding of any single block of audio is inhibited if AQM > $AQM_{THRESH+}$ for that block of audio, and wherein $AQM_{THRESH+}$ > AQM_{THRESH}.

17. The method of claim 14 wherein each of the y blocks of audio has a spectral energy, wherein

$$aqm[b] = \frac{|E_C[b]-E_0[b]|}{E_{MASK}[b]}$$

wherein each of the y blocks of audio is divided into B frequency bands, wherein b is a variable representing a frequency band within the B frequency bands, wherein $E_0[b]$ is the spectral energy in a frequency band b of each of the blocks of audio without coding, wherein $E_C[b]$ is the spectral energy in the frequency band b of each of the blocks of audio with coding, and wherein

$$AQM = \sum_{b=0}^{b=B} aqm[b] \; .$$

18. The method of claim 1 wherein the first block of audio has energy, wherein step b) is inhibited if E_1/E_2 < E_{PRE}, wherein E_1 is the energy in a first portion of the first block of audio, wherein E_2 is the energy in a second portion

of the first block of audio, and wherein E_{PRE} is a predetermined reference.

19. The method of claim 18 wherein E_1 represents the energy in a first quarter of the first block of audio and E_2 represents the energy in a last three quarters of the first block of audio.

20. The method of claim 18 wherein the first block of audio is sampled to produce S samples, wherein E_1 is determined according to the following equation:

$$E_1 = \sum_{s=0}^{s=d} A^2[s]$$

wherein E_2 is determined according to the following equation:

$$E_2 = \sum_{s=d+1}^{s=S} A^2[s]$$

wherein A[s] is the amplitude A of a sample s in the first block of audio, and wherein d is chosen so as the divide E_1 from E_2.

21. The method of claim 20 wherein d is chosen so that E_1 represents the energy in a first quarter of the first

block of audio and so that E_2 represents the energy in a last three quarters of the first block of audio.

22. The method of claim 1 further comprising the step of encoding l blocks of audio with binary code bits, wherein n of the binary code bits are arranged to designate a distribution level of encoding.

23. The method of claim 22 wherein k groups of l blocks of audio in an audio stream are set aside for encoding by k distributors of the audio, and wherein a predetermined combination of the n binary code bits of a corresponding group of l blocks of audio is arranged to indicate that the corresponding group of l blocks of audio has not been encoded by a distributor.

24. The method of claim 1 further comprising the step of encoding more than one frequency band within the first and second blocks of audio with corresponding binary code bits, wherein each frequency band is encoded by a different distributor of the audio.

25. The method of claim 1 further comprising the following steps:

e) encoding, by use of a primary encoder, a group of blocks of audio with a synchronization sequence, wherein the primary encoder leaves a predetermined number of groups of additional blocks of audio uncoded; and,

f) encoding, by use of either the primary encoder or a secondary encoder, a first corresponding one of the groups of blocks of additional audio so as to identify a first distributor of the audio; and,

g) encoding, by use of a secondary encoder, a second corresponding one of the groups of additional blocks of audio so as to identify a second distributor of the audio.

26. A method for encoding a block of audio with a binary code bit comprising the following steps:

a) selecting a frequency from a frequency spectrum of the block of audio;

b) selectively amplifying an odd index frequency in a neighborhood of the selected frequency to be a local maximum if the block of audio is to be encoded with the binary code bit having a first value; and,

c) selectively amplifying an even index frequency in a neighborhood of the selected frequency to be a local maximum if the block of audio is to be encoded with the binary code bit having a second value.

27. The method of claim 26 wherein the block of audio is a first block of audio, wherein the binary code bit is a first binary code bit, wherein the frequency selected in step a) is a first frequency, and wherein the method further comprises the following steps:

d) selecting a second frequency from a frequency spectrum of a second block of audio, wherein the second frequency bears a predetermined offset relationship to the first frequency;

e) selectively amplifying an odd index frequency in a neighborhood of the second frequency to be a local maximum if the second block of audio is to be encoded with a second binary code bit having one of the first and second values; and,

f) selectively amplifying an even index frequency in a neighborhood of the second frequency to be a local maximum if the second block of audio is to be encoded with the second binary code bit having the other of the first and second values.

28. A method for encoding blocks of audio with binary code bits comprising the following steps:

a) determining an audio quality measure AQM for each block of audio;

b) comparing the AQM corresponding to each block of audio to AQM_{THRESH}, wherein AQM_{THRESH} is a predetermined audio quality measure reference;

c) if $AQM < AQM_{THRESH}$ for x blocks of audio out of y blocks of audio, encoding the blocks of audio with binary bits, wherein x and y are corresponding predetermined numbers of blocks of audio; and,

d) if $AQM > AQM_{THRESH}$ for the x blocks of audio out of the y blocks of audio, suspending encoding of the blocks of audio.

29. The method of claim 28 wherein y is a predetermined number of consecutive blocks of audio.

30. The method of claim 28 wherein encoding of any single block of audio is inhibited if $AQM > AQM_{THRESH+}$ for that block of audio, and wherein $AQM_{THRESH+} > AQM_{THRESH}$.

31. The method of claim 28 wherein each of the blocks of audio has a spectral energy, wherein

$$aqm[b] = \frac{|E_C[b] - E_0[b]|}{E_{MASK}[b]}$$

wherein each of the blocks of audio is divided into B frequency bands, wherein b is a variable representing a frequency band within the B frequency bands, wherein $E_0[b]$ is the spectral energy in a frequency band b of each of the blocks of audio without coding, wherein $E_C[b]$ is the spectral energy in the frequency band b of each of the blocks of audio with coding, and wherein

$$AQM = \sum_{b=0}^{b=B} aqm[b] \ .$$

32. The method of claim 31 wherein y is a predetermined number of consecutive blocks of audio.

33. The method of claim 28 wherein step c) comprises the steps of increasing the spectral power at one of first and second frequencies within a corresponding block of audio so as to render the spectral power at the one of the first and second frequencies a local maximum, and decreasing the spectral power at the other of the first and second frequencies so as to render the spectral power at the other of the first and second frequencies a local minimum.

34. The method of claim 28 wherein step c) comprises the step of selectively changing a phase relationship between first and second frequencies, and wherein the first and second frequencies are within a block of audio.

35. The method of claim 28 wherein step c) comprises the steps of swapping a spectral amplitude at a first frequency with a spectral amplitude at a second frequency, wherein the first frequency is within an encoded block of audio, and wherein the second frequency has a maximum amplitude in a frequency neighborhood of the first frequency.

36. A method of encoding a block of audio with a binary code bit, wherein the block of audio has an energy, and wherein the method comprises the following steps:

a) determining a ratio E_1/E_2, wherein E_1 is the energy in a first portion of the block of audio, and wherein E_2 is the energy in a second portion of the block of audio;

b) modulating the block of audio with the binary code bit if $E_1/E_2 > E_{PRE}$, wherein E_{PRE} is a predetermined reference; and

c) not modulating the block of audio with the binary code bit if $E_1/E_2 < E_{PRE}$.

37. The method of claim 36 wherein E_1 is the energy in a first quarter of the block of audio, and wherein E_2 is the energy in a last three quarters of the block of audio.

38. The method of claim 36 wherein the block of audio is sampled to produce S samples, wherein E_1 is determined according to the following equation:

$$E_1 = \sum_{s=0}^{s=d} A^2[s]$$

wherein E_2 is determined according to the following equation:

$$E_2 = \sum_{s=d+1}^{s=S} A^2[s]$$

wherein A[s] is the amplitude A of a sample s, and wherein d is chosen so as the divide E_1 from E_2.

39. The method of claim 38 wherein d is chosen so that E_1 is the energy in a first quarter of the block of audio and so that E_2 is the energy in a last three quarters of the block of audio.

40. The method of claim 36 wherein step b) comprises the steps of increasing the spectral power at one of first and second frequencies within the block of audio so as to render the spectral power at the one of the first and second frequencies a local maximum, and decreasing the spectral power at the other of the first and second frequencies within the block of audio so as to render the spectral power at the other of the first and second frequencies a local minimum.

41. The method of claim 36 wherein step b) comprises the step of selectively changing a phase relationship between first and second frequencies, and wherein the first and second frequencies are within the block of audio.

42. The method of claim 36 wherein step b) comprises the steps of swapping a spectral amplitude at a first frequency with a spectral amplitude at a second frequency, wherein the first frequency is within the block of audio, and wherein the second frequency has a maximum amplitude in a frequency neighborhood of the first frequency.

43. A method of encoding blocks of audio with binary code bits comprising the following steps:

a) encoding each of the blocks of audio with a binary bit by modulating the audio within the corresponding block of audio at selected first and second frequencies, wherein the selected first and second frequencies are hopped from block to block; and,

b) executing step a) so as to indicate first and second levels of distribution of the audio.

44. The method of claim 43 wherein step a) comprises the step of encoding l blocks of audio with binary code bits such that n of the binary code bits are arranged to designate a distribution level of encoding.

45. The method of claim 44 wherein k groups of the l blocks of audio in an audio stream are set aside for encoding by k distributors of the audio, and wherein a predetermined combination of the n binary code bits of a corresponding group of l blocks of audio is arranged to indicate that the corresponding group of l blocks of audio has not been encoded by a distributor.

46. The method of claim 43 wherein step a) comprises the step of encoding more than one frequency band within each block of audio with corresponding binary code bits, and wherein each frequency band is encoded by a different distributor of the audio.

47. The method of claim 43 further comprising the following steps:

e) encoding, by use of a primary encoder, a first group of blocks of audio with a synchronization sequence, wherein the primary encoder leaves a at least second and third groups of audio blocks uncoded;

f) encoding a second group of blocks of audio so as to identify a first distributor of the audio; and,

g) encoding a third group of blocks of audio so as to identify a second distributor of the audio.

48. The method of claim 47 wherein the first and second groups of blocks of audio are the same group of blocks of audio.

49. The method of claim 47 wherein the second group of blocks of audio is encoded by the primary encoder.

50. The method of claim 43 wherein step a) comprises the following steps:

increasing the spectral power at one of the first and second frequencies of each block of audio so as to render the spectral power at the one of the first and second frequencies a local maximum; and,

decreasing the spectral power at the other of the first and second frequencies of each block of audio so as to render the spectral power at the other of the first and second frequencies a local minimum.

51. The method of claim 43 wherein step a) comprises the step of selectively changing a phase relationship between the first and second frequencies in each of the blocks of audio.

52. The method of claim 43 wherein step a) comprises the following step swapping, in each of the blocks of audio, a spectral amplitude of at least one of the first and second frequencies with a spectral amplitude of a frequency having a maximum amplitude in a frequency neighborhood of the at least one of the first and second frequencies.

53. The method of claim 43 wherein the hopping is performed in accordance with a predetermined frequency hop sequence.

54. The method of claim 43 wherein the hopping is based on low frequency maxima.

55. The method of claim 43 wherein a synchronization block is added to the signal, and wherein the synchronization block is characterized by a triple tone portion.

56. A method for decoding first and second blocks of audio in order to recover corresponding first and second binary code bits therefrom comprising the following steps:

a) detecting first and second frequencies from a frequency spectrum of the first block of audio;

b) demodulating the first and second frequencies in order to recover to the first binary code bit;

c) detecting third and fourth frequencies from a frequency spectrum of the second block of audio, wherein the third and fourth frequencies bear a predetermined offset relationship to the first and second frequencies; and,

d) demodulating the third and fourth frequencies in order to recover the second binary code bit.

57. The method of claim 56 wherein:

step b) comprises the step of demodulating the first and second frequencies to produce a binary code bit having a value dependent upon which of the first and second frequencies is a local maximum and which of the first and second frequencies is a local minimum; and,

step d) comprises the step demodulating the third and fourth frequencies to produce a binary code bit having a value dependent upon which of the third and fourth frequencies is a local maximum and which of the third and fourth frequencies is a local minimum.

58. The method of claim 56 wherein:

step b) comprises the step of demodulating the first and second frequencies depending upon a phase relationship between the first and second frequencies; and,

step d) comprises the step of demodulating the third and fourth frequencies depending upon a phase relationship between the third and fourth frequencies.

59. The method of claim 56 wherein:

step b) comprises the step of demodulating the first and second frequencies based upon a swapping of a spectral amplitude of at least one of the first and second frequencies with a spectral amplitude of a frequency having a maximum

amplitude in a frequency neighborhood of the least one of the first and second frequencies; and,

step d) comprises the step of demodulating the third and fourth frequencies based upon a swapping of a spectral amplitude of at least one of the third and fourth frequencies with a spectral amplitude of a frequency having a maximum amplitude in a frequency neighborhood of the least one of the third and fourth frequencies.

60. The method of claim 56 wherein the predetermined offset relationship is determined by a frequency hop sequence.

61. The method of claim 56 wherein the predetermined offset relationship is determined by frequency hopping based on a low frequency maximum.

62. The method of claim 56 wherein the predetermined offset relationship is determined in accordance with a reference frequency, a frequency hop sequence, and a predetermined shift index.

63. The method of claim 62 wherein the first and second frequencies are determined according to the following equations:

$$I_1 = I_{5k} + H_S - I_{shift}$$

and

$$I_0 = I_{5k} + H_S + I_{shift}$$

wherein I_1 and I_0 are indexes corresponding to the first and second frequencies respectively, wherein I_{5k} is an index corresponding to the reference frequency, wherein H_S is a component of the frequency hop sequence, and wherein I_{shift} is the predetermined shift index.

64. The method of claim 56 wherein the predetermined offset relationship is determined according to a reference frequency, a low frequency maximum, and a predetermined shift index.

65. The method of claim 64 wherein the first and second frequencies are determined according to the following equations:

$$I_1 = I_{5k} + I_{\max} - I_{shift}$$

and

$$I_0 = I_{5k} + I_{\max} + I_{shift}$$

wherein I_1 and I_0 are indexes corresponding to the first and second frequencies respectively, wherein I_{5k} is an index corresponding to the reference frequency, wherein I_{max} is an index corresponding to a frequency at which the spectral power of the first block of audio is a maximum in a low frequency band of the first block of audio, and wherein I_{shift} is the predetermined shift index.

66. The method of claim 56 further comprising the step of detecting a synchronization message from a plurality of blocks of audio, and wherein the synchronization message is characterized by a triple tone portion.

67. The method of claim 56 further comprising the following steps:

decoding l blocks of audio to recover a plurality of binary code bits; and,

decoding n of the plurality of binary code bits in order to determine a distribution level of the audio.

68. The method of claim 67 wherein a particular combination of the n binary code bits indicates that a corresponding group of 1 blocks of audio has not been encoded by a distributor of the audio.

69. The method of claim 56 further comprising the step of decoding more than one frequency band within the first and second blocks of audio in order to recover corresponding pairs of first and second binary code bits, wherein each frequency band is encoded by a different distributor of the audio.

70. The method of claim 56 further comprising the following steps:

e) decoding a first group of blocks of audio so as to identify a first distributor of the audio; and,

f) decoding a second group of blocks of audio so as to identify a second distributor of the audio.

71. A method of decoding blocks of audio encoded with binary code bits comprising the following steps:

a) decoding each of the blocks of audio in order to recover a corresponding binary bit by demodulating the audio within the corresponding block of audio at selected first and second frequencies, wherein the selected first and second frequencies are hopped from block to block; and,

b) executing step a) so as identify first and second distributors of the audio.

72. The method of claim 71 wherein step a) comprises the step of decoding n of the binary code bits in order to determine a distribution level of encoding.

73. The method of claim 72 wherein k groups of blocks of audio in an audio stream are set aside for encoding by k distributors of the audio, and wherein step a) comprises the step of decoding a predetermined combination of the n binary code bits to determine that a corresponding group of blocks of audio has not been encoded by a distributor.

74. The method of claim 71 wherein step a) comprises the step of decoding more than one frequency band within each block of audio in order to recover a corresponding binary code bit from each frequency band and to identify a different distributor of the audio associated with each frequency band.

75. The method of claim 71 further comprising the following steps:

e) decoding a first group of blocks of audio to recover a synchronization sequence;

f) decoding a second group of blocks of audio so as to identify a first distributor of the audio; and,

g) decoding a third group of blocks of audio so as to identify a second distributor of the audio.

76. The method of claim 75 wherein the first and second groups of blocks of audio are the same group of blocks of audio.

77. The method of claim 71 wherein step a) comprises the step of demodulating the first and second frequencies to recover a binary code bit having a value dependent upon which of the first and second frequencies is a local

maximum and which of the first and second frequencies is a local minimum.

78. The method of claim 71 wherein step a) comprises the step of demodulating the first and second frequencies to recover a binary code bit having a value dependent upon a phase relationship between the first and second frequencies.

79. The method of claim 71 wherein step a) comprises the step of demodulating the first and second frequencies based upon a swapping of a spectral amplitude of at least one of the first and second frequencies with a spectral amplitude of a frequency having a maximum amplitude in a frequency neighborhood of the least one of the first and second frequencies in order to recover a single binary code bit.

80. The method of claim 71 wherein the hopping is performed in accordance with a predetermined frequency hop sequence.

81. The method of claim 71 wherein the hopping is based on low frequency maxima.

82. The method of claim 71 further comprising the step of decoding a synchronization message characterized by a triple tone portion.

83. A method of decoding a block of audio in order to recover a binary code bit therefrom comprising the following steps:

a) detecting a frequency having an amplitude maximum within a selected frequency neighborhood of the block of audio;

b) if the frequency detected in step a) corresponds to an odd frequency index, decoding the frequency as a binary code bit having a first value; and,

c) if the frequency detected in step a) corresponds to an even frequency index, decoding the frequency as a binary code bit having a second value.

84. The method of claim 83 wherein a frequency index bears a predetermined relationship to a corresponding frequency.

85. The method of claim 84 wherein the predetermined relationship is given by the following equation:

$$I = (\frac{k}{j}) \cdot f.$$

86. The method of claim 85 wherein k = 255 and wherein j = 24.

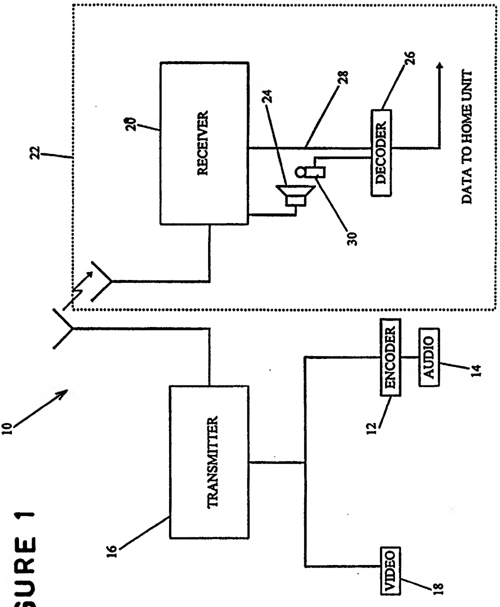

10
22
20
RECEIVER
24
28
26
DECODER
30
DATA TO HOME UNIT
TRANSMITTER
16
ENCODER
12
AUDIO
14
VIDEO
18

FIGURE 1

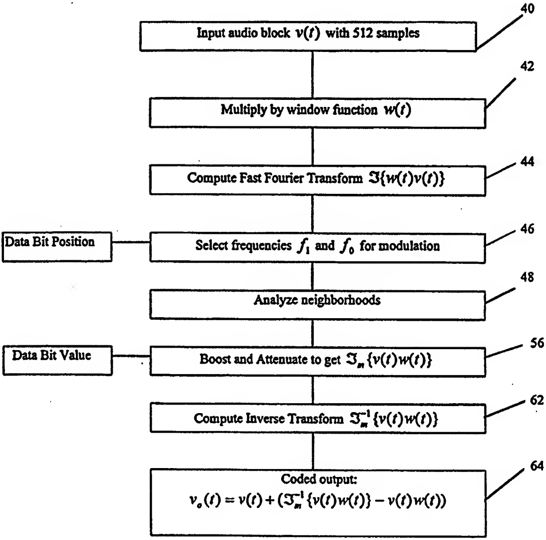

Input audio block $v(t)$ with 512 samples
40
Multiply by window function $w(t)$
42
Compute Fast Fourier Transform $\Im\{w(t)v(t)\}$
44
Data Bit Position
Select frequencies f_1 and f_0 for modulation
46
Analyze neighborhoods
48
Data Bit Value
Boost and Attenuate to get $\Im_m\{v(t)w(t)\}$
56
Compute Inverse Transform $\Im_m^{-1}\{v(t)w(t)\}$
62
Coded output:
$v_o(t) = v(t) + (\Im_m^{-1}\{v(t)w(t)\} - v(t)w(t))$
64

FIGURE 2

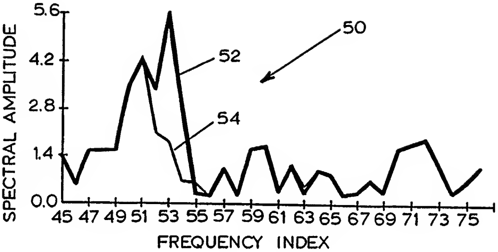

5.6
4.2
2.8
1.4
0.0
SPECTRAL AMPLITUDE
50
52
54
45 47 49 51 53 55 57 59 61 63 65 67 69 71 73 75
FREQUENCY INDEX

FIGURE 3

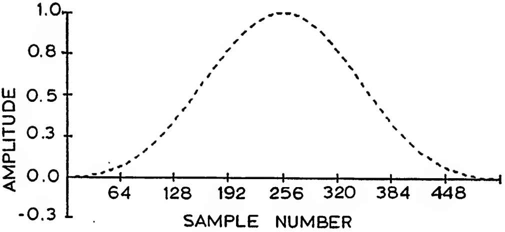

1.0
0.8
0.5
0.3
0.0
-0.3
AMPLITUDE
64
128
192
256
320
384
448
SAMPLE NUMBER

FIGURE 4

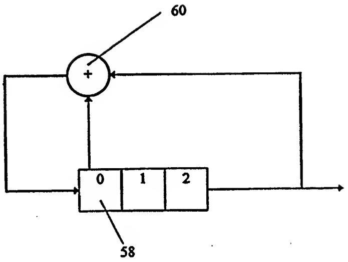


FIGURE 5

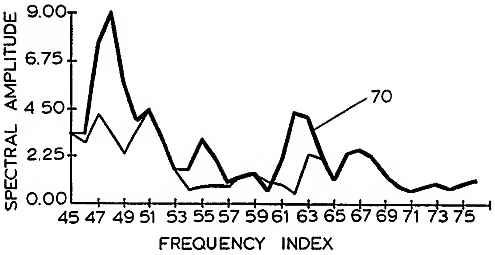

SPECTRAL AMPLITUDE
9.00
6.75
4.50
2.25
0.00
70
45 47 49 51 53 55 57 59 61 63 65 67 69 71 73 75
FREQUENCY INDEX

FIGURE 6

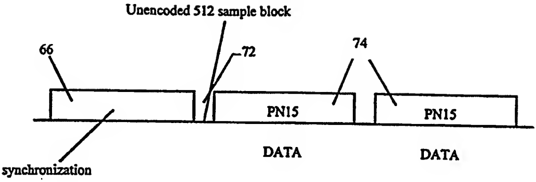


FIGURE 7A

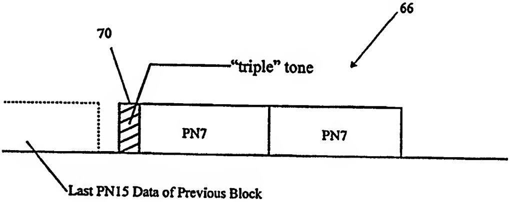
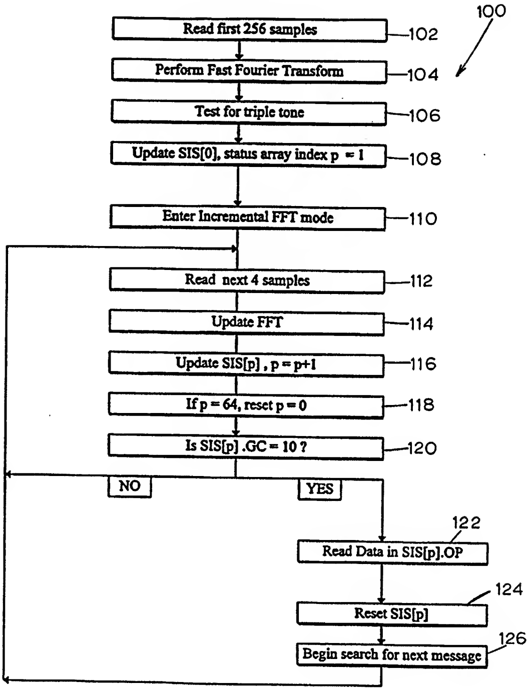


FIGURE 7B



100
Read first 256 samples
102
Perform Fast Fourier Transform
104
Test for triple tone
106
Update SIS[0], status array index p = 1
108
Enter Incremental FFT mode
110
Read next 4 samples
112
Update FFT
114
Update SIS[p] , p = p+1
116
If p = 64, reset p = 0
118
Is SIS[p] .GC = 10 ?
120
NO
YES
122
Read Data in SIS[p].OP
124
Reset SIS[p]
126
Begin search for next message

FIGURE 8

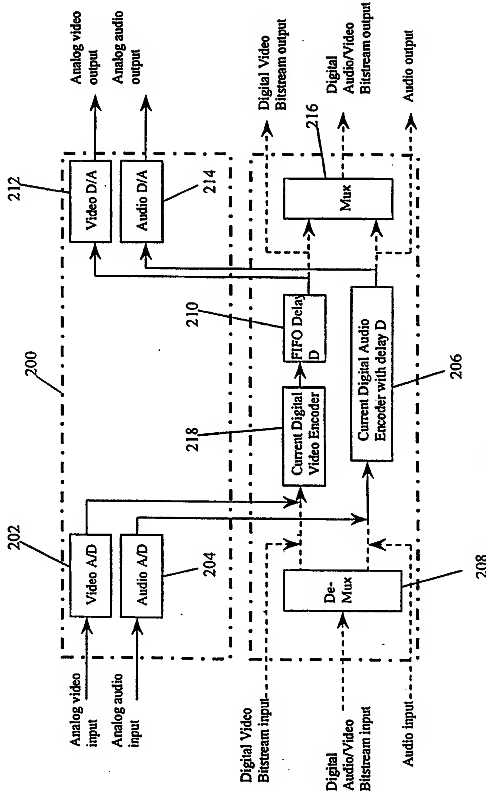

Analog video input
Analog audio input
Digital Video Bitstream input
Digital Audio/Video Bitstream input
Audio input
200
202
204
Video A/D
Audio A/D
De-Mux
208
218
Current Digital Video Encoder
Current Digital Audio Encoder with delay D
206
210
FIFO Delay D
212
Video D/A
Audio D/A
214
Mux
216
Analog video output
Analog audio output
Digital Video Bitstream output
Digital Audio/Video Bitstream output
Audio output

FIGURE 9

INTERNATIONAL SEARCH REPORT

Int onal Application No
PCT/US 00/03829

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04H1/00 H04H9/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04H

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the International search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, PAJ, INSPEC

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
P,X	WO 00 04662 A (NIELSEN MEDIA RESEARCH INC) 27 January 2000 (2000-01-27)	1-11,22, 23,26, 27, 43-45, 50-68, 70-73, 77-86
P,Y	the whole document ---	12,14-16
X	GB 2 292 506 A (ARBITRON COMPANY THE) 21 February 1996 (1996-02-21)	28-30
Y	page 4, line 5-17 page 15, line 10-20; claims 1-4,12-14; figures 1,3 --- -/--	12,14-16

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

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"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
3 August 2000	18/08/2000

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Pantelakis, P

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
X	GB 2 260 246 A (ARBITRON COMPANY THE) 7 April 1993 (1993-04-07) page 4, line 5-17 page 6, line 4-13 page 12, line 10-20; claims 1-3,12-14,16; figures 1,3	28-30
A	US 4 972 471 A (DALE JAMES ET AL) 20 November 1990 (1990-11-20) column 1, line 46 -column 2, line 26; claims 1-4,12-15,17,18; figure 1	1,26,28, 36,43, 56,71,83

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			AT	174737	T	15-01-1999
			AU	668888	B	23-05-1996
			AU	2668692	A	03-05-1993
			DE	69227916	D	28-01-1999
			DE	69227916	T	27-05-1999
			EP	0606341	A	20-07-1994
			EP	0872971	A	21-10-1998
			ES	2125907	T	16-03-1999
			GB	2260246	A,B	07-04-1993
			GR	3029415	T	28-05-1999
			HK	1007850	A	23-04-1999
			MX	9205506	A	01-07-1993
			WO	9307689	A	15-04-1993
			US	5574962	A	12-11-1996
			US	5581800	A	03-12-1996
			US	5787334	A	28-07-1998
			ZA	9207317	A	24-03-1994
GB 2260246	A	07-04-1993	FR	2681997	A	02-04-1993
			AT	174737	T	15-01-1999
			AU	668888	B	23-05-1996
			AU	2668692	A	03-05-1993
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			DE	69227916	T	27-05-1999
			EP	0606341	A	20-07-1994
			EP	0872971	A	21-10-1998
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			GB	2292506	A,B	21-02-1996
			GR	3029415	T	28-05-1999
			HK	1007850	A	23-04-1999
			MX	9205506	A	01-07-1993
			WO	9307689	A	15-04-1993
			US	5574962	A	12-11-1996
			US	5581800	A	03-12-1996
			US	5787334	A	28-07-1998
			ZA	9207317	A	24-03-1994
US 4972471	A	20-11-1990	CA	2088780	A	15-03-1992
			WO	9205550	A	02-04-1992
			EP	0548071	A	30-06-1993